REMARKS

In view of the comments which follow, and pursuant to 37 C.F.R. § 1.111, reconsideration of the Official Action of December 5, 2003 is respectfully requested by Applicants.

Summary

Claims 1 – 18 stand rejected, and are pending following entry of this response.

Rejections under 35 U.S.C. § 102

The Examiner has rejected claims 1, 2, 8, 9, 12, 13 and 16 under 35 U.S.C. § 102 (b) as being anticipated by Arslan et al. (Arslan) (U.S. Patent No.: 6,263,307). Applicants respectfully traverse these rejections.

The pending claim 1 is directed to a voice feature extraction device that comprises a noise reduction system coefficient calculation unit that calculates beforehand a noise reduction system coefficient, and an input voice power spectrum calculation unit that calculates a power spectrum vector of a processed input voice, so that a noise reduction system set to the calculated coefficient processes the power spectrum vector of the processed input voice.

Applicants submit that pending claim 1 is patentable over the cited art, because Arslan does not disclose either the beforehand calculation of the noise reduction coefficient or the extraction of the voice feature in the form of a power spectrum vector.

In regard to the beforehand calculation of the noise reduction coefficient, the reference Arslan discloses that the filter coefficients are generated in noise filter block 208 (column 5, lines 65 – 66, Figure 2), and further that the filter coefficients in block 208 derive from estimates for the noise spectrum and the noisy speech spectrum of the frame (column 6, lines 3 – 5). Therefore, the filter coefficients are generated subsequent to the Fast Fourier Transform (FFT) module 204 in Figure 2

and also subsequent to the FFT module 902 as in Figure 9a and 9b, which convert a time domain input speech signal into a frequency domain signal. Thus, Arslan calculates the filter coefficient as the FFT communicates the frequency domain converted frames to the multiplier, wherein the filter coefficient is pointwise multiplied by converted frames. In contrast, a noise reduction system coefficient calculation unit separate from the noise reduction system calculates Applicants' noise reduction system coefficient, before it is communicated to the noise reduction system. The noise reduction system is then set to the noise reduction coefficient and executes an operation processing a power spectrum vector.

In regard to the extraction of the voice feature, as shown in Figures 9a and 9b, the Arslan reference discloses that the time domain input speech signal is transformed into a frequency domain signal via the FFT module 902, then a multiplier 920 performs the noise suppression of the frequency domain signal. Further, an inverse FFT (IFFT) 922 transforms the noise suppressed frequency domain signal into a time domain signal and outputs it. In addition, although an input voice power spectrum is calculated by a magnitude squarer 904, the input voice spectrum is only used to calculate the coefficient to set the multiplier 920 which performs the noise suppression calculation. Therefore, in Arslan the voice extraction feature is performed via a calculation about a voice signal. In contrast, Applicants' voice feature extraction is performed via a calculation about a voice power spectrum, i.e., the voice feature is extracted by the noise reduction system in the form of a power spectrum vector from the input voice power spectrum vector generated by the power spectrum calculation unit.

Hence, for at least the above discussed reasons, claim 1 is not anticipated by Arslan. Claim 2 is dependent on claim 1 and is likewise not anticipated by Arslan.

Applicants also submit that pending claim 8 likewise is patentable over the cited art, because Arslan does not disclose either the beforehand calculation of the noise reduction coefficient, or the extraction of the voice feature in the form of a power spectrum vector. As in the above claim 1 discussion, Applicants respectfully

submit that claim 8 is not anticipated by Arslan. Claim 9 is dependent on claim 8 and is likewise not anticipated by Arslan.

Further, Applicants submit that pending claim 12 likewise is patentable over the prior art, because Arslan does not disclose either a step of calculating in advance the noise reduction filter coefficient, or the extraction of the voice feature in the form of a power spectrum vector. As claim 12 is directed to a method of extracting voice features from a noisy speech input signal, Applicants respectfully submit that as shown in the above claim 1 discussion, Arslan's step of calculating the filter coefficient is not performed beforehand, but rather as the FFT communicates the frequency domain converted frames to the multiplier, wherein the filter coefficient is pointwise multiplied by converted frames. Hence, claim 12 is not anticipated by Arslan. Claim 13 is dependent on claim 12 and is likewise not anticipated by Arslan.

Still further, Applicants also submit that pending claim 16 similarly is patentable over the cited art, because Arslan does not disclose either a step of calculating in advance the noise reduction coefficient or a step of calculating a voice feature from a power spectrum vector. As in the above claim 12 discussion, Applicants respectfully submit that claim 16 is not anticipated by Arslan.

Accordingly, Applicants therefore respectfully request that the rejections of claims 1, 2, 8, 9, 12, 13 and 16 under 35 U.S.C. § 102(b) be withdrawn.

Rejections under 35 U.S.C. § 103

The Examiner has next rejected claims 3, 10, 14 and 17 under 35 U.S.C. § 103 (a) as being unpatentable over Arslan et al. (Arslan) (U.S. 6,263,307) in view of Im et al. (Im) (US 5,805,696). Applicants respectfully traverse these rejections.

The pending claim 3, dependent on claim 2, recites that the filter coefficient calculation unit executes an adaptive control to a signal having an input voice signal and a simulated voice signal added, and obtains a tap coefficient to thereby calculate the filter coefficient.

Applicants submit that pending claim 3 is patentable over the cited art, because Arslan does not disclose the beforehand calculation of the noise reduction coefficient feature and the extraction of the voice feature in the form of a power spectrum vector feature, both of claim 2 and independent claim 1, and because Im teaches away from claim 3, in that the adaptive filter of Im receives a control signal combining incoming signals and a training signal is obtained as their algebraic difference rather than their algebraic sum.

In regard to Im, an adder 28 provides to an adaptive estimating filter 20 a control signal equal to the algebraic difference between a training signal P(z) and a resulting signal T(z) of all incoming signals. In contrast, Applicants' filter coefficient calculation unit executes an adaptive control to a signal having an input voice signal and a simulated voice signal added.

Hence, as shown in the above discussion of the 102(b) rejections, Arslan does not disclose all the limitations of claim 2, and consequently fails to disclose the corresponding limitations of claim 3. Further, Im also fails to teach an adaptive filter having the input voice signal and the simulated voice signal added as input. Consequently, the two prior art references may not be combined to reject claim 3 under 35 U.S.C. 103(a).

Claims 10, 14, and 17 are, directly or indirectly, dependent on claims 8, 12, and 16, respectively. Since Arslan does not disclose all the limitations of claims 8, 12, and 16, Arslan consequently fails to disclose the corresponding limitations of claims 10, 14, and 17. Further, claims 10, 14 and 17 include the feature of adding the input voice signal to the simulated voice signal as input to the adaptive filter. Applicants have shown in relation to claim 3 that Im fails to teach this feature. Consequently, the Arslan and Im references may not be combined to reject claims 10, 14, and 17 under 35 U.S.C. 103(a). Therefore, Applicants respectfully request that the rejections of 3, 10, 14 and 17 be withdrawn.

The Examiner has further rejected claims 5 - 7 under 35 U.S.C. §103 (a) as being unpatentable over Arslan et al. (Arslan) (U.S. 6,263,307) in view of LaRue (US 5,805,696). Applicants respectfully traverse these rejections.

Claims 5 - 7 are dependent on claim 1. Applicants submit that although the Larue reference is directed to a voice feature extraction device applied to a vehicle navigational system, Arslan has been shown to fail to disclose all the limitations of claim 1, so likewise Arslan fails to disclose the corresponding limitations of claims 5 - 7. Therefore, Arslan may not be combined with LaRue to reject claims 5 - 7 under 35 U.S.C. 103(a). Thus, Applicants respectfully request that the rejections of 5 - 7 be withdrawn.

The Examiner has further rejected claims 4, 11, 15 and 18 under 35 U.S.C. §103 (a) as being unpatentable over Arslan et al. (Arslan) (U.S. 6,263,307) in view of Im et al. (Im) (US 5,805,696) and further in view of Haykin et al. (Haykin) (U.S. 5,027,123). Applicants respectfully traverse these rejections.

Claims 4, 11 and 15 are either directly or indirectly dependent on claims 1, 8 and 12, respectively. Arslan has been shown to fail to disclose all the limitations of claims 1, 8 and 12, so Arslan likewise fails to disclose the corresponding limitations of claims 4, 11 and 15. In addition, Im has been shown to fail to disclose the limitations of claims 3, 10 and 14, so likewise Im fails to disclose the corresponding limitations of claims 4, 11 and 15. Therefore, Arslan, Im and Haykin may not be combined to reject claims 4, 11, and 15 under 35 U.S.C. 103(a).

As for claim 18, Arslan fails to disclose an FFT operation unit that executes a fast Fourier transform to a filter coefficient obtained by an adaptive control of the adaptive filter. In Arslan, the filter coefficients are generated in noise filter block 208 (column 5, lines 65 – 66, Figure 2), and further the filter coefficients in block 208 derive from estimates for the noise spectrum and the noisy speech spectrum of the frame (column 6, lines 3 – 5). Therefore, the filter coefficients are generated subsequent to the Fast Fourier Transform (FFT) module 204 as in Figure 2, and

also subsequent to the FFT module 902 as in Figures 9a and 9b. To the contrary, Applicants' filter coefficient is calculated before being supplied to an FFT operation unit; hence Arslan does not disclose at least this feature of claim18.

Further as shown above, Im fails to teach an adaptive filter having the input voice signal and the simulated voice signal added as input. Thus, Applicants submit that since both Arslan and Im have been shown above as failing to disclose the corresponding limitations of claim 18, then Arslan, Im and Haykin may not be combined to reject claim 18 under 35 U.S.C. 103(a).

As such, Applicants respectfully request that the rejections of 4, 11, 15 and 18 be withdrawn.

Conclusion

Applicants submit that this application is now in condition for allowance, and favorable reconsideration of this application in view of the above remarks is respectfully requested. Allowance of claims 1 - 18 at an early date is earnestly solicited.

If the Examiner finds that there are any outstanding issues which may be resolved by a telephone interview, the Examiner is invited to contact the undersigned attorney at the below listed number.

Respectfully submitted,
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